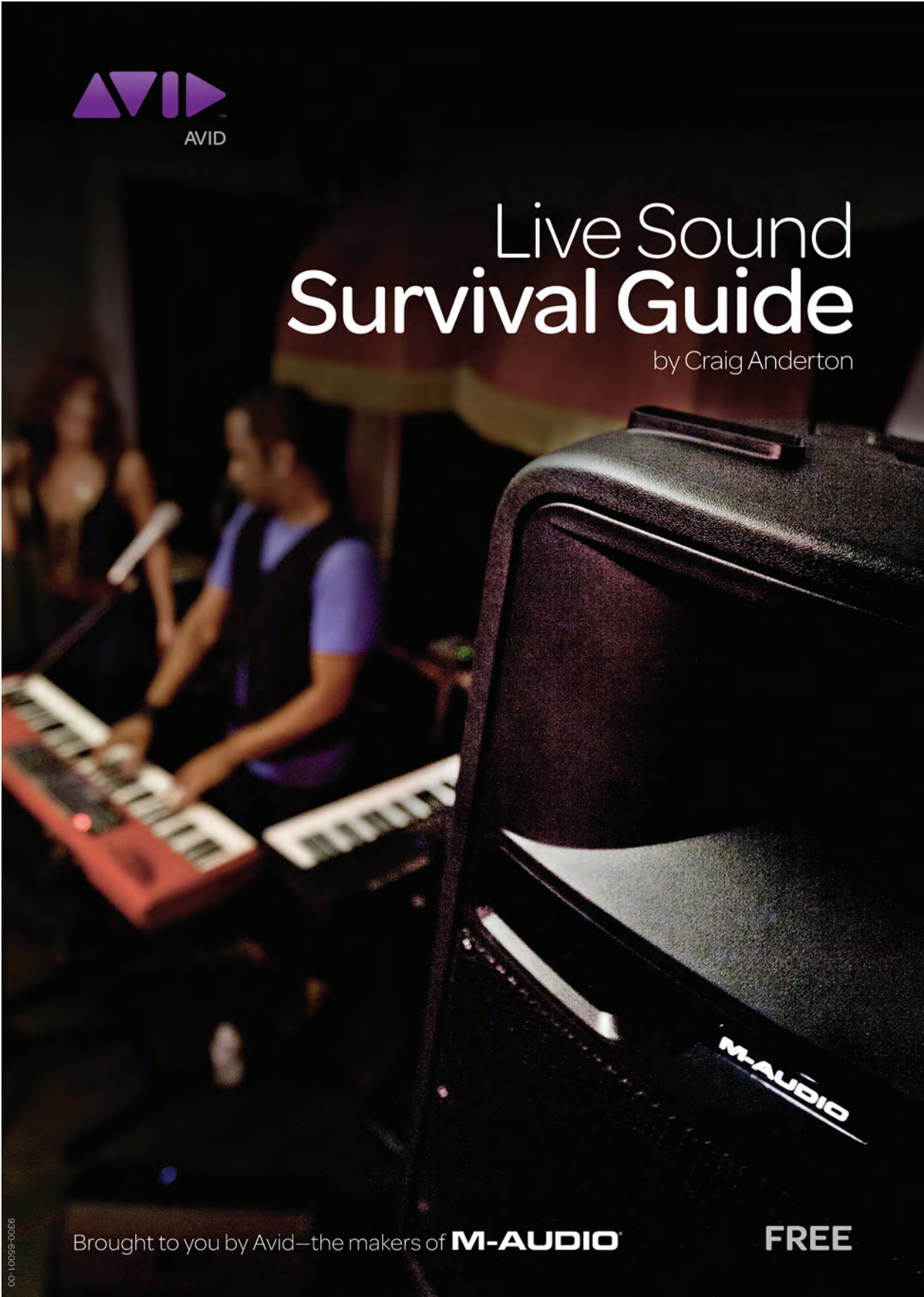
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Production Staff

Editor-in-Chief
Craig Anderton

Editors
John Krogh
Michael Parker

Advertising Contact
Laurie Blondin
626-610-2434

Art Direction and Layout
Patrick Wong

Photography
Theo Jemison
Joshua Merrill
Craig Anderton

CHAPTER 1:

PA System Basics and Components

"PA" stands for "Public Address," and the key word here is "public." You want to amplify a band, presenter, auctioneer, lecturer, conference, worship service, or other sound source so that the audio can be heard clearly by a large group of people–your listening public.

The most common public address system components are:

Loudspeaker. Loudspeakers convert electrical energy into acoustic energy–moving air that we can hear with our ears. In addition to loudspeakers that are like home hi-fi speakers on steroids, public address systems often include *subwoofers.* These are speakers optimized to reproduce bass, as bass requires more power and different speaker construction than higher audio frequencies. Speakers are built into *cabinets,* which can often mount on speaker stands, and include handles to simplify transportation and setup.

Power amplifier. A power amplifier receives incoming audio signals, and increases their power so they can drive the speakers. Amplifier power is measured in watts. The higher the wattage, the louder the potential levels you can achieve with your system, although of course the speakers need to be able to handle the available power. In modern, compact systems like the M-Audio® GSR series, the amplifiers are built into the same cabinet as the speakers. As a result, all three elements–speaker, amplifier, and cabinet–are optimized to work together efficiently. In addition, this kind of system is more transportable (with fewer wiring issues) because it's more self-contained.

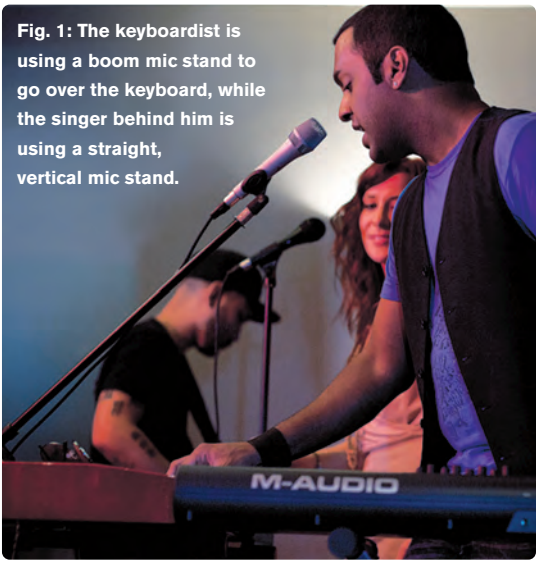


Fig. 1: The keyboardist is using a boom mic stand to go over the keyboard, while the singer behind him is using a straight, vertical mic stand.

Mixer. A mixer combines multiple signal sources, typically microphones and instrument outputs, into a single, unified output that can then feed the power amplifiers. However, not all situations require a mixer. The M-Audio GSR10 and GSR12 speakers have microphone and instrument inputs–so for simple setups, it's possible to feed a mic or instrument directly into the GSR, and not require any other connections.

Cables. Although placing an amp and speaker within a single cabinet eliminates the need to connect these two elements with a cable, you still need cables to connect the mixer to the amplifier, and input signals to the mixer. Although you don't need to buy ultra-expensive audiophile cables, quality cables are important for reliability, and it's crucial to always carry spares as backup.

Microphones. These are the mirror image of speakers, as they convert acoustical energy

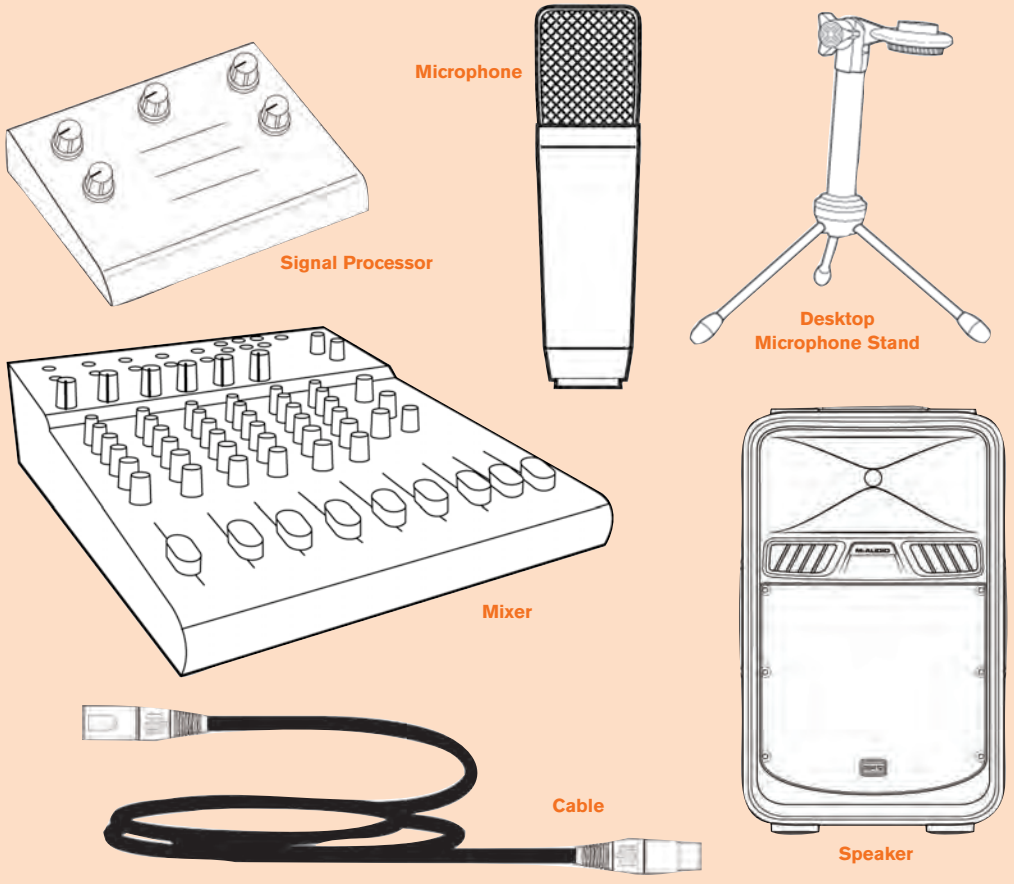
(such as a vocalist or instrument sound) into electrical energy that can feed the mixer or power amplifier input. Microphone quality relates directly to overall sound quality, but fortunately, microphone prices have come down over the years so you don't have to spend much to get good quality. It's also possible to use wireless microphones that transmit sound as radio frequencies; a receiver picks up this signal, converts it to audio, then feeds your mixer. Wireless microphones help minimize clutter on stage, and are ideal for people who like to move around stage while performing.

Microphone stands. If you're not using wireless microphones, you'll need a place to "park" the mic. Even if a presenter likes to hand-hold the mic, at some point the mic will need to be put somewhere. There are two main types of mic stands: straight and boom (Fig. 1). Boom stands enable you to position the mic away from the stand if the stand gets into the way of, for example, an instrumentalist.

Signal processors. These devices alter the basic audio signal coming from a mic or instrument to enhance the tone or quality, as well as compensate for deficiencies in either the equipment or the audio source. For example, if a lecturer's voice has a thin quality, signal processors can make the voice sound fuller and deeper.

The room and audience. You may not think of them as part of a PA system, but as we'll see later, they can have a major effect on the overall sound.

PA SYSTEM COMPONENTS



CHAPTER 2:

Choosing and Using the Right Microphones

In the studio or on stage, the mic is the first link in the audio chain. We'll look at which mic types work best for live use, then describe how to get the best sound from them.

MICROPHONE TYPES

No matter which mic you choose, two key factors are ruggedness and directionality (the ability to pick up a particular sound source without picking up others).

There are two popular mic types for PA systems (Fig. 2).

Dynamic microphones are physically rugged and can handle high sound pressure sound levels, so they're the most common choice for PA systems. They also resist noise from handling, making them popular in hand-held applications (e.g., vocalists). The tradeoff is that the sound isn't as refined as other technologies, but live, these differences are negligible.

Condenser microphones are common for recording due to their excellent frequency response and ability to respond to *transients* (rapid changes in level, as from percussion). They're more fragile than dynamic mics, and most models need a power supply—either from an internal battery, or from "phantom" power that can be supplied by all but the least expensive mixers (see Chapter 4 on Mixers). However, condenser mics designed for live use are getting more rugged, and with proper care, can hold up to the rigors of the road. As they're sensitive to handling noise, they're usually mounted on mic stands.

There are two common condenser mic types. *Small-diaphragm* mics are more sensitive, so they excel at reproducing transients. *Large-diaphragm* mics tend to give a "warmer" sound, and are often used for vocals in the studio.

Fig. 2: The mic on the left is the M-Audio SoundCheck, an affordable dynamic microphone. The one on the right is the Nova, a condenser mic that's also from M-Audio.

A third type of mic technology, the *ribbon* microphone, is seldom used live due to fragility.

MIC PLACEMENT AND USAGE

Any mic needs to be placed so that it picks up sound optimally—but to do that, you need to know some mic basics.

Mic Directionality

This is crucial for live use, as you want mics to pick up specific sounds and discriminate against other sounds (e.g., you don't want a vocal mic to pick up instrument sounds on the same stage). Different mics have different *pickup patterns* for specific applications, sort of like cameras—some "see" only a small part of what we can see with our eyes, but there are also cameras with wide-

angle lenses, panorama cameras, etc. See Fig. 3 for pickup pattern diagrams.

Cardioid (unidirectional) mics pick up only those sounds in front of the mic. Thus the mic can "aim" at a vocalist or instrumentalist. Popular examples of M-Audio microphones are the SoundCheck (a rugged, unidirectional dynamic), Nova® (a large-diaphragm condenser), and Pulsar II (a small-diaphragm type designed for studio use, but suitable for live).

Hypercardioid (also called *supercardioid*) mics have an even narrower sound field. They're a good choice for vocalists fronting loud bands.

Omnidirectional mics pick up sounds from all directions. They can be useful with conferences, where you have many people speaking but may not have a mic for each person, and volume levels aren't particularly loud.

Figure 8 mics tend to be ribbon types, which as noted, are rarely used live. However, some condenser mics can also achieve this response.

Mic Placement Issues

The *proximity effect* is most pronounced with dynamic cardioid types–as you get closer to the mic, the apparent bass response rises. Singers with good mic technique use this to their advantage: On soft, intimate parts they'll hold the mic close to the mouth, to exaggerate the bass and give a warmer sound. Acoustic guitars can also use the proximity effect for a more bass-heavy sound. In any event, either use it to your advantage or compensate for it.

The *inverse square law* states that the sound level drops dramatically the further the mic is from the sound source. This can be a problem with vocalists who lack good mic technique, alternately "swallowing" the mic and then backing away, without changing their volume to compensate.

The mic's *angle* in relation to the sound source affects tone. Generally, aiming the mic's element at the sound source results in the brightest

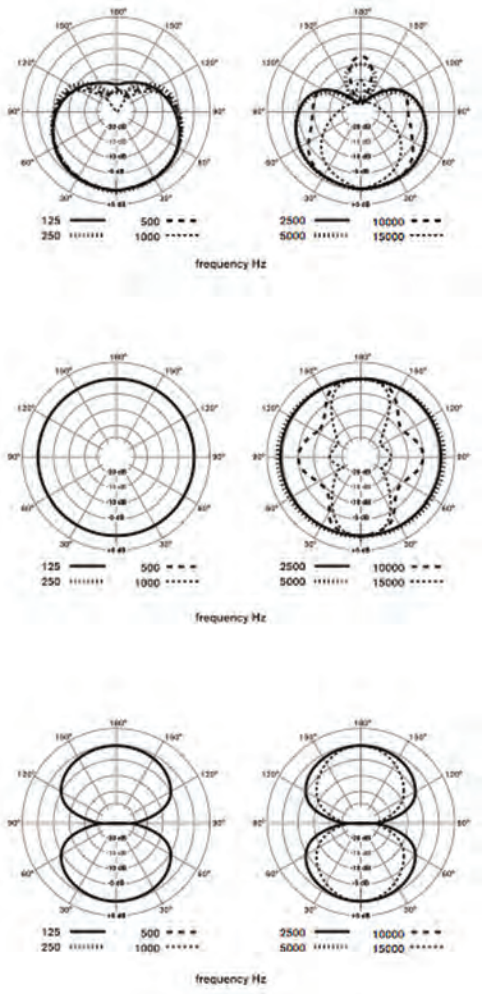


Fig. 3: The top polar pattern shows a cardioid response; note how it rejects sounds from the rear. The middle pattern shows an omnidirectional response, which picks up sounds from all directions. The bottom pattern shows a figure-eight response, so-called because it picks up sounds from the front and back of the mic.

sound (be careful, though; with some mics the "entrance" is at the top, and with others, at the side–check your mic's documentation). Having the mic at an angle produces a somewhat mellower, less "direct" sound.

Understanding these three issues helps you decide how to mic particular sound sources. For example, soft sound sources require having the mic up close, but you may need to decrease the bass response (see Chapter 6 on signal processing, as well as the section on

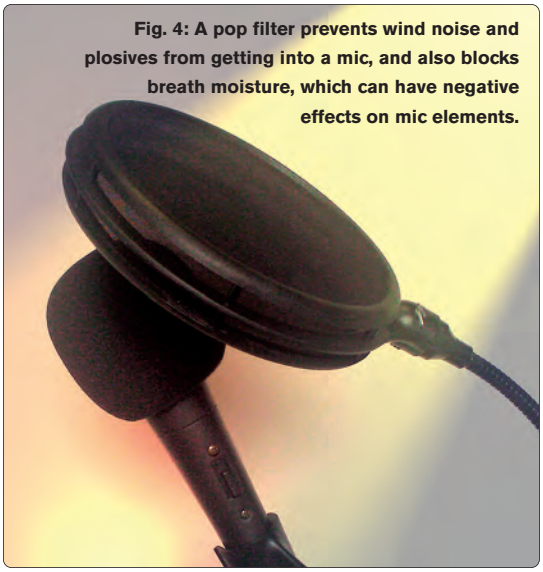


Fig. 4: A pop filter prevents wind noise and plosives from getting into a mic, and also blocks breath moisture, which can have negative effects on mic elements.

Optimizing the Sound toward the end of this chapter) to compensate for the proximity effect.

With louder sound sources like wind instruments, you can back the mic away a bit, and there won't be major volume variations if the player moves around a bit. When miking something like a guitar amp, you have a lot of latitude on mic placement–the standard is to point it at the middle of the speaker in a speaker cabinet, but different placements can yield different tones.

OPTIMIZING THE SOUND

Now that you've chosen your mics and set them up, here are several ways to optimize the sound.

Mic Switches

Mics often include switches for tailoring the sound, such as:

- ***On/off (mute) switch.*** This is handy when, for example, handing a mic from one person to another, or briefly turning off the mic if a sneeze or cough is imminent.
- ***Low-cut filter.*** These reduce low-frequency content. If a vocalist's or lecturer's "b" and "p" sounds cause a loud "popping," or the proximity effect is a problem, turn the filter on.
- ***Pad switch.*** This reduces the mic's sensitivity. For example, a loud guitar amp might be so loud it overloads a mixer's mic input. Enabling the pad switch will minimize distortion.

Reducing Feedback

Feedback occurs when a signal from the speaker gets into the mic, and therefore gets re-amplified. Fixing feedback involves several elements of any PA system: Speaker placement, mixer control settings, and mics. Using directional mics that point away from speakers, and avoiding the proximity effect are two ways to reduce the possibility of feedback at the mic itself. For more information, see Chapter 9 on Troubleshooting.

Dealing with "Thin" Sounds

If you're using two mics with an instrument like piano, you can run into *phasing* problems if each mic picks up a different portion of the sound wave. This can result in a "thin," unnatural sound. If your mixer has a Phase switch for each channel, try flipping this for one of the mics in the pair. If this solves the problem, fine. Otherwise, for more information on how to solve this problem, refer to Chapter 9.

Reducing Wind and Breath Noise

The rush of air from sounds like "b" and "p" can produce nasty pops. Also, wind noise can create similar problems. You can use a cover made of acoustic foam that slips over the mic's head and reduces these sounds–or use a clip-on nylon mesh screen filter between the vocalist and the mic (Fig. 4).

Extending Microphone Life

Abused mics may continue to function, but lose sound quality. Always store mics in their protective cases.

CABLES AND CONNECTORS

Cables connect the various PA elements, providing a way for signals to get from one place to another. In the next chapter, we'll cover how to choose the right cable, as well as get the best performance out of your cables.

CHAPTER 3:

Balanced/Unbalanced Lines and Connectors

Cables for unbalanced lines have two conductors, while balanced lines have three conductors (don't confuse this with typical stereo cables, which also have three conductors; balanced lines are for carrying a single, monophonic signal). Balanced lines can help reject interference from sources like fluorescent light buzzes, hum, and even radio frequencies from passing mobile transmitters (e.g., taxis, CB radio)–as long as they connect to inputs and outputs whose circuitry is designed to take advantage of balanced lines. This technology is particularly useful with low-level signals, as the interference might be almost as strong as the signal itself.

Unbalanced lines are more common, and include guitar cords as well as most hi-fi cables. Unbalanced cables don't reject interference as well, but they're generally less expensive, and work fine in applications with strong signal levels.

The two different types of lines can use different connectors. Fig. 5 shows an unbalanced 1/4" phone plug; note that it has two sections, one for each line. This is also called a TS type of plug because it has a "tip" and a "sleeve" section.

Fig. 6 shows a balanced 1/4" phone plug. These are called TRS (tip-ring-sleeve) connectors because they have three sections: The tip and sleeve (like a standard unbalanced phone connector) but also, a third "ring" section in between the tip and the sleeve.

Fig. 7 shows an XLR plug, as used for balanced connections. The three pins connect to the three conductors.

As for jacks, you can't tell whether 1/4" phone jacks are designed for balanced or unbalanced lines from the outside. However, most equipment manufacturers will label their jacks (Fig. 8) so you know what type they are. XLR jacks (Fig. 9) are balanced; you can see three holes for the three plug pins.

A relatively recent type of jack, the combination jack (Fig. 10), gets its name because it combines XLR and 1/4" phone jack capabilities–you can insert either type of plug (but of course, not both at the same time). With 1/4" phone connections, the combo jack can work with balanced or unbalanced lines.

Another unbalanced connector, the phono connector (Fig. 11), is more common with consumer gear. However, some PA mixers include RCA phono jacks for one or two inputs in case you need to interface with something like a portable CD player. (Phono connectors are also used for S/PDIF digital signals.)

Here are guidelines about which type of gear uses which type of line and connector.

Microphones. Almost all mics for PA applications use balanced lines and XLR connectors.

Synthesizers and other electric instruments. Many of these devices offer balanced line outputs that work with balanced or unbalanced lines. Unless the cable run is very long or there are audible sources of interference, unbalanced lines will work fine.

Turntable preamps, portable CD players, etc. These consumer devices typically use unbalanced lines with RCA phono

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Fig. 5: An unbalanced 1/4" phone plug. Note the tip, the black insulating band, and ground. The flare toward the base of the jack provides a stronger grip when inserted into a jack.

Fig. 6: A balanced (TRS) 1/4" phone plug. Note there are three sections, separated by black insulating bands–the tip, the ring, and the ground. These are also called stereo plugs when used to carry two independent left and right channel signals.

Fig. 7: An XLR plug, as used for balanced line connections.

connections. Since cable runs are usually short, unbalanced lines are not a problem.

Portable MP3 players. These use stereo mini-jack and plug connectors, which resemble 1/4" phone types but are smaller (Fig. 12). Most pro audio gear does not accept this type of connector, so you if you want to use one, you'll likely need an adapter that converts the stereo mini-jack output to two separate, unbalanced lines, typically terminated in 1/4" phone or phono plugs.

Electric guitar and bass. Almost all use unbalanced lines.

Mixer inputs. The mic inputs will accept balanced lines with XLR jacks. The line ins usually accept either balanced or unbalanced lines via 1/4" jacks, but some higher-end gear uses XLR jacks for balanced line-level signals.

Powered speakers. Models like the M-Audio GSR series powered speakers accept the line-level signals coming from today's mixers and include combo jacks, so they can interface with whatever connector your mixer uses.

IMPEDANCE: NO WORRIES!

With today's PA systems, *impedance* (which, to simplify greatly, represents the "friction"

Fig. 8: The M-Audio DMP3 preamp has output jacks toward the left, but looking at the jacks, you wouldn't know whether they're balanced or unbalanced. So, they're labeled as being balanced.

Fig. 9: The XLR jack is the large, round connector toward the left. Also note the red LED to its lower left, which indicates that phantom power to the mic is enabled.

Fig. 10: The combination XLR jack is toward the left, and accepts both XLR and 1/4" phone plugs.

audio signals encounter at inputs and outputs, and is measured in ohms) isn't something you need to know much about–just follow a few basic guidelines.

Microphones. Almost all PA microphones have a low-impedance output and generate low-level signals. So, connect low-impedance mic outputs to low-impedance mixer mic inputs (Chapter 4) as these inputs are designed specifically to accept low-level, low-impedance signals.

Synthesizers, turntable preamps, portable music players, etc. These invariably have low-impedance inputs and generate high-level ("line-level") signals. Impedance matching is not an issue if you simply plug their outputs into mixer inputs that handle line levels.

Electric guitar and bass. These are special cases, because the output impedances are too high for mic inputs, while the signal levels are too high for mic inputs but too low for line-level inputs. Some mixers have special guitar inputs; otherwise, if you plan to plug a guitar into a PA system directly, you'll probably need to use a preamp to match impedance and boost the level. However, if the guitar goes through various effects, a multi-effects processor, or a pedalboard, these will almost certainly generate enough level to feed line-level mixer inputs.

Powered speakers. Models like the M-Audio GSR series aren't very sensitive to impedance, so simply plugging the mixer output into the speaker's input will do the job. With older, non-powered speakers, the rules regarding impedance are pretty simple too: Match the speaker impedance with the same impedance output on your powered mixer or power amp. For example, if an amp has 4- and 8-ohm outputs, and the speaker is rated at 8 ohms, plug it into the 8-ohm output.

CABLES

Here are tips on the care and handling of cables.

- Assess what kind of cables you need to connect your various pieces of gear. Don't buy longer cables just because they'll work

Fig. 11: The M-Audio Fast Track Pro audio interface offers balanced TRS 1/4" phone jack outputs, but note the four RCA phono output connectors to the left of the TRS output jacks. The inset shows an RCA phono plug.

for shorter runs too—have a selection of short and long cables, then use the appropriate length.

- Have plenty of spare cables as backup. Cables are generally reliable, but can fail (and always seem to fail at the most inopportune time).
- Use balanced lines with mics.
- For relatively short cable runs—under 20-25 feet or so—unbalanced lines are fine for outputs from musical instruments like keyboards, direct outputs from guitar and bass amps, and the like. But, there's certainly no harm in using balanced lines if the gear accommodates them.
- Always unplug cables by grasping the plug, *never* by pulling on the wire.
- Avoid sharp cable bends, stepping on cables, rolling heavy objects over them, and running them near power lines or heat sources.
- Where cables are exposed to the public, tape them to the floor with non-residue gaffer's tape. Make sure people can't trip over them.
- Carry cable ties (available at office supply stores) to bundle cables together for a neater setup. However, don't bundle cables carrying strong signals (like speaker connections) with cables carrying low-level signals (like mics), as there may be crosstalk.
- The metal in plugs and jacks can oxidize over time, degrading the connection. Squirt some contact cleaner (available at stores like Radio Shack, or a more pro product like Caig's DeoxIT) on the plug, then plug and unplug a couple dozen times. This will remove the oxidation and improve the contact.
- Connect all your cables *before* turning on power to your PA system. If you must connect or disconnect a cable while power is on, turn off the amplifier or powered speakers, then turn them back on again when all connections are made.
- Always have some adapters on hand, like XLR to 1/4". You never know when to expect the unexpected.

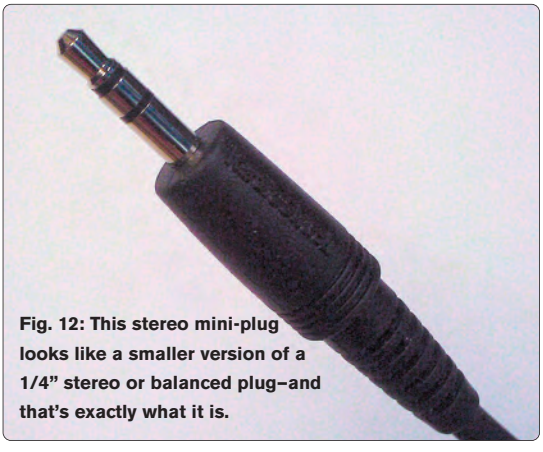

Fig. 12: This stereo mini-plug looks like a smaller version of a 1/4" stereo or balanced plug—and that's exactly what it is.

CHAPTER 4:

Speaker, Mixer, and Monitor Setup

Where you put the speakers, mixer, mics, and performers has a huge impact on the audience experience as well as the effectiveness of the PA system itself. We'll look at the optimum placement for these elements, but first, let's consider the proper order for turning on pieces of gear.

The most important point to remember is to turn on the amplifier (or powered speaker systems) *last,* after all connections have been made. Plugging and unplugging devices while the system is on can make loud pops that are bad for your speakers and bad for your ears. Likewise, *always* power-up your system with the mixer's master volume control (or powered speaker level controls) at zero–full off. Turn up the gain only when you know that all is well.

SPEAKER POSITIONING

The primary rule for speaker positioning is to place (from the audience's viewpoint) any speakers in front of the mics used by performers or presenters, with as much distance as possible between the mics and speakers. This allows for the maximum gain before feedback. One exception is DJ setups if a microphone is not a major part of the performance. In this case, speakers can be on a horizontal line with the DJ or even slightly behind, and if the DJ has a wireless mic, he can step behind the speaker for announcements.

When mounted on speaker mounting tripods, speakers should be higher than the audience and tightened down so they cannot move or rotate on the tripods. If someone walks in front of the speaker, the speaker itself should be higher than the person; in some venues,

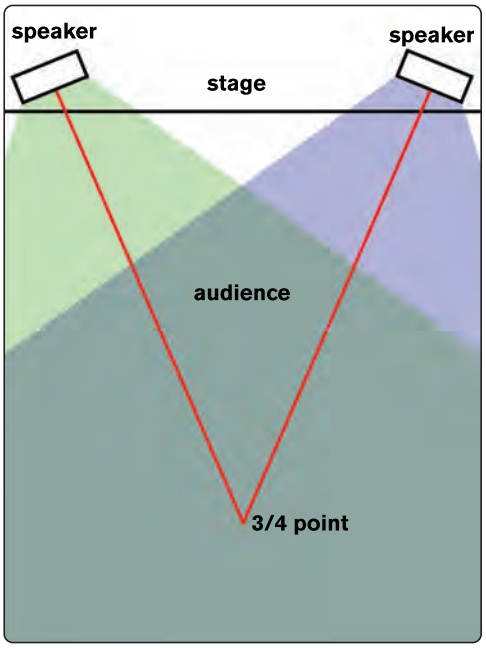

Fig. 13: In a small auditorium, angling the speakers slightly inward toward the audience can improve coverage. You may even be able to get away with using only one speaker.

mounting the speakers even higher provides better coverage.

The GSR10 and GSR12 speakers allow for stacking up to two speakers vertically, using an interlock system that fastens the top of one speaker to the bottom of an identical speaker model. The extra height can improve coverage dramatically. To avoid tip-overs, make sure the combination is stable and out the way of the audience and performers.

To provide the best coverage with two speakers, mount them toward the left and right sides of the stage, and angle them slightly

toward the middle–draw an imaginary line down the middle of the audience, and aim the speakers to a point about three-quarters of the way toward the far end of the room (Fig. 13). Note that high frequencies emanating from a loudspeaker are highly directional, whereas low frequencies can bend around objects in a room. If everyone in the audience can see the speaker(s), they'll likely be able to hear the high frequencies. This also means that subwoofers (like the GSR18) do not need to be raised, and in fact, will sound better if they sit on the floor or the stage (which is fortunate, as subwoofers are heavy). Furthermore, special pole mounts are available for mounting a GSR10 or GSR12 above the GSR18, providing an optimized high-frequency/low-frequency speaker pair.

SPEAKERS AND WALLS

The audience doesn't hear only the sound from the speakers, but sounds reflected off the walls and other objects. If there's a significant time difference between these two sound sources, music becomes less distinct, and presenters less intelligible.

Sound travels at about one foot per millisecond (a thousandth of a second, or ms). While delays under about 30ms are generally not perceived as objectionable, delays over 30ms produce an echo effect. Ideally, for an audience member the difference in distance between the speaker and the nearest reflective hard surface (which will produce the loudest reflection) will be 30 feet or less. Of course this is not always possible, but the closer you can come to this ideal, the better. Also note that the longer the reflection path, the weaker the sound. So even though a reflection might be more than 30 feet away, if it has to travel a considerable distance (especially if there are absorptive surfaces in the room, like people, clothing, drapes, etc.) it won't have as much influence.

Sounds bouncing off a side wall tend to be less problematic than sounds bouncing off a wall behind the speaker, or in front of it. Referring to Fig. 14, person "A" in the back of the hall will hear the direct sound from the speaker, and the

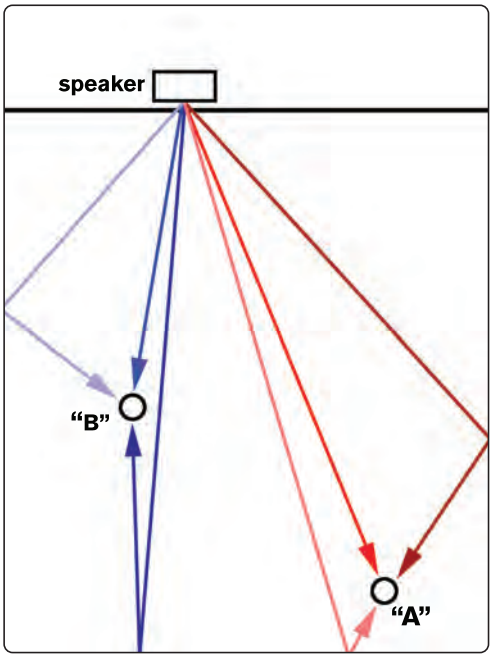

Fig. 14: Person "A" may actually hear better sound quality than person "B," who's closer to the speaker, depending on a variety of factors.

reflection from the back wall. This person will hear reasonably good sound quality because the difference between these two sound sources will be less than 30 feet, the angled sounds coming off the wall won't be that different in length than the direct sound, and the rear reflection will be much weaker anyway. On the other hand, person "B" will hear the sound from the speaker, long side reflections from the walls, and a reflection from the back of the room–which may be a problem if the back wall is hard and reflects a lot of the sound's energy. This is one reason why speakers are angled toward the middle-to-rear of the audience, as the more direct sound will be louder than the reflections and minimize any echo effect.

POSITIONING THE MIXER

At professional concerts, the person mixing the sound is usually stationed toward the center-middle of the hall, in order to hear what the audience hears. This is not always practical in other situations, especially since long cable runs (called "snakes") are necessary to connect the mixer to the mics on stage, as well as to the speakers at the side of the stage.

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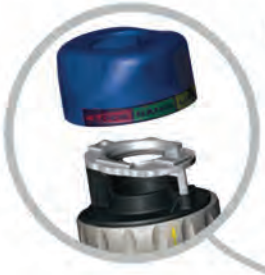

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A common approach for smaller venues (e.g., conference rooms in convention centers) is to position the mixer off to one side, fairly close to one of the speakers. The person doing the mixing will hear the direct speaker sound; while this doesn't take room reflections into account, the balance among sound sources for those hearing the direct speaker sound will be correct (or at least, be what the mixer hears).

In situations where the person doing the mixing should not be visible for aesthetic reasons (such as plays), or there is no room in front of the stage for a mixer, one option is to have the mixer off to the side of the stage, just behind the edge of a curtain. The mixer will need to rely on headphones and during sound check, confirm that what's being heard in the headphones is representative of what the audience is hearing.

POSITIONING PERFORMERS OR PRESENTERS

As mentioned, mics should be behind the speakers to minimize feedback. When performers use mic stands, they can be positioned in place. Wireless mics are more of a problem, particularly with lead singers who like to "work the crowd," or motivational speakers who roam the stage. These people need to be told beforehand to avoid moving in front of the speakers, and whoever is mixing should watch them closely and reduce the gain if it appears that their movement might result in feedback. Hypercardioid mics work well in this context, and the person should speak or sing close to the mic so that the voice's level is much louder than any sound that might be coming from the speakers.

POSITIONING MONITORS

The performers will not hear themselves accurately since they are standing behind the speakers. For some people, like presenters, this doesn't really matter. For singers, it can be crucial because they may not be able to stay on pitch if they can't hear themselves.

In-ear reference earphones, which resemble earbuds, are generally considered to be an optimum solution. Otherwise, it's common to put a "wedge" monitor speaker in front of a performer, pointing up to them at a 45-degree angle (Fig. 15). Using hypercardioid mics minimizes pickup of the sound coming from these speakers, and the volume levels needn't be high as the speakers are pointing directly at the performer from a short distance away. Both the GSR10 and GSR12 speakers can function as wedge monitor speakers when laid on their sides. The GSR10 is ideal for this application due to its smaller size, making it less obtrusive to the audience.

TONAL SETUP

Adjusting the overall tone to match a particular venue can improve the sound quality dramatically. One common solution is to use an equalizer (either one built in to the mixer, or added in the signal path between mixer and amplifier), but the M-Audio GSR10 and GSR12 speakers take a simpler approach by offering four "tuning" modes that optimize the speakers for specific types of applications and rooms. *Normal* mode is the standard listening mode, with flat response. *Hi-Fi* gives more presence for music playback in an application like a school dance or when providing background music. *DJ* mode boosts the lows and gives more definition, thus providing a "DJ sound" even at relatively low volumes.

Voice mode is optimized when plugging a mic directly into the GSR10 and GSR12 for conferences, boardroom meetings, seminars, and other applications where all you really need is an amplified voice (no music) and aren't using a mixer.

Fig. 15: If leaned on their sides, the GSR10 and GSR12 can serve as floor wedge monitors.

CHAPTER 5:

Mixers

The mixer is your PA's traffic director–signals enter from various sources, and are routed to one or more sets of outputs that feed the speakers. Mixers may look daunting, but they consist primarily of multiple, identical channels: Learn one channel, and you've learned the others too.

INPUTS AND OUTPUTS

The main mixer specification is the number of inputs and outputs. For example, an 8-in, 2-out mixer can accept eight different signal sources, and combine them into two outputs (also called buses). These two outputs can provide stereo (left and right) channels.

You can think of the input signal path as a vertical, downward flow into the mixer, and the bus (output signal path) as a horizontal flow from left to right out of the mixer (Fig. 16). Typical mixers have anywhere from four to dozens of inputs.

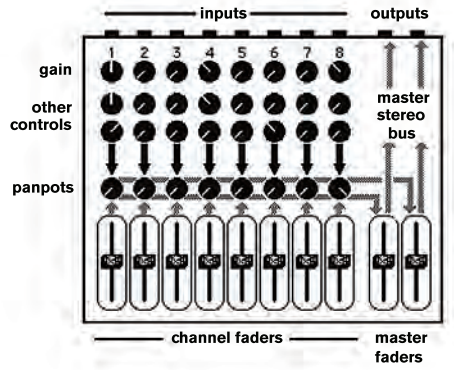

Fig. 16: Signals flow from the inputs through the mixer channels, to the channel faders, then to the panpots. At this point, the signals go to a master stereo bus, which flows through the master faders, then to the outputs.

CHANNEL STRIPS

Each mixer channel has its own channel strip (also called an input module) for processing or routing a signal before sending it to the output. A basic channel strip (Fig. 17) includes some or all of the following:

Input jack(s). This can be a low-impedance XLR mic input, a line-level input for stronger signals, both types of inputs with a selector switch, a stereo signal input, or other, less common options. Mixers often include different inputs–a 16-input mixer might have eight XLR mic inputs, and eight line-level 1/4" inputs.

Insert jack. This allows patching external gear, such as particular equalizers or compressors (see Chapter 6) into the mixer channel's signal path. To save space, the insert jack is usually a TRS (tip-ring-sleeve) type, and requires a special breakout cable to feed a signal processor's input and output.

Preamplifier. Intended to amplify low-level signals (e.g., mic outputs), the preamp will offer large amounts of gain, as set by a *gain control*. If the channel strip also offers a line input, the gain control will be bypassed, or cover a different, more appropriate gain range. Preamps help bring all external signals to the same approximate level prior to being mixed together with the faders (described later).

Phantom power switch. When on, this sends +48V to an XLR mic input to power a condenser mic.

Low-cut filter. Although not always included in mixers, this reduces low frequencies to minimize room rumble, mic "pops," etc. Similarly, a **high-cut filter** reduces high frequencies to lessen sibilance and hiss.

Fig. 17: Here's a typical channel strip, from the M-Audio NRV10 mixer/interface. From top to bottom, there's an XLR mic input, a 1/4" line input for balanced or unbalanced lines (which you can use instead of the XLR in), TRS insert jack, mic/line switch to tailor the gain control characteristics to the chosen input, preamp gain control, three EQ controls for high/mid/low frequencies, two aux send controls (one to a monitor out, one to the onboard digital effects), pan control, and level fader. Note the red clipping indicator to the upper right of the level fader, along with a yellow light to indicate when a channel is muted, as chosen by the Mute switch at the bottom of the channel strip.

Clipping indicator. This is usually an LED that lights if the signal exceeds the preamp's available dynamic range, which means you should turn down the preamp gain. Some inputs also have an "activity" LED that lights if there's a signal present (which verifies that a signal is reaching the mixer channel). Another variation combines both functions into a bi-color LED that glows green to indicate an activity, and red to indicate overload.

Equalizer. This is a type of tone control; see Chapter 6.

Send control. See the section on Mixer Buses.

Solo switch. Soloing a channel mutes all other input modules. This is useful for hearing exactly what's happening in a channel without being distracted by the other channels.

Mute switch. Muting a channel is the same as pulling the fader all the way down. Use mute switches to keep channels out of the mix until right before they're needed.

Panpot. With stereo mixers, this places the input signal anywhere in the stereo field–from full left to full right.

Fader. This sets the level of the channel in relation to the other channels. It may be a linear slider (Fig. 18), or a rotary control.

MIXER BUSES

Basic mixers have two output buses. More advanced models have several buses (called *send* or *auxiliary* buses), and you can set up different mixes on these different buses using send or aux level controls. These "pick off" part of a channel strip's signal, and send it to a bus in the same way a fader regulates the level going to the master stereo output (Fig. 19).

One common use for a mixed bus output is to set up a separate mix for singers. This is because if the singers can hear themselves coming from the main speakers, then that sound is probably getting into their mics, which encourages feedback. But setting up the mics behind the speakers to minimize feedback makes it difficult for the singers to hear themselves, making it tough to stay on pitch. As mentioned previously, one solution is to provide the singers with reference earphones. A separate mix can be set up specifically for the singer using an auxiliary bus, perhaps with vocals and melodic instruments up loud, and drums turned down. This aux bus output feeds a separate headphone amp, which drives the reference earphones. Thus the singers can hear themselves without having the monitor signal get into their mics.

Another bus application is to provide a master volume control for several individual channels. For example, suppose you're mixing a church choir with multiple mics. You can send a signal from each mic to a bus, turn down the mic channel faders, then return the bus output to a

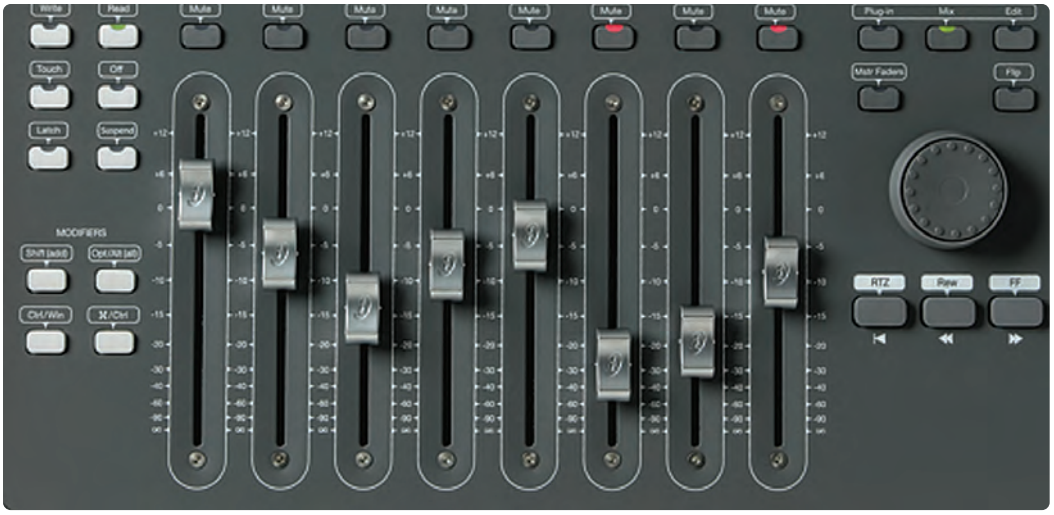

Fig. 18: Linear faders are very easy to adjust, particularly because you can move more than one at a time. These eight linear faders are part of the Avid 003 interface for Pro Tools recording software.

mixer input or send (aux) return input (Fig. 20) to blend it back in with the main output. Use the send controls to balance each mic perfectly, then if you want to raise or lower the entire choir, use the mixer input level or send return control rather than adjusting each send control individually.

Send controls usually have a pre/post switch to pick up the pre-channel fader signal (so the level remains constant regardless of the fader setting) or post-channel fader, so that pulling down the fader pulls down the send to the bus as well.

Each mixer has its own way of sending signals to buses. If space is tight and there are lots of buses, a channel might have a send control along with switches that route the send control to a particular bus (or buses).

THE MASTER SECTION

A mixer's master section includes "global" controls, such as:

Master output control. This varies the mixer's overall output level.

Send bus master output controls. While not present on all mixers, these affect the overall level of the bus they control. In the example of the singer listening through reference earphones, the bus master control would adjust the level going to the reference earphones.

Bus return controls. In the choir application mentioned above, you may not need to use up a mixer input for the bus return, as there may be some "return" inputs specifically for this purpose. These will be line level and generally not offer all the options of a typical channel strip, but usually include a level control. They can also be used as extra inputs in a pinch.

Output meters. Most mixers have at least an output meter, and high-end mixers might include meters for individual channels. A typical meter will consist of 10 or more LEDs, and use different colors to differentiate among ranges of levels (green for normal, yellow for close to overload, and red for overload). Avoid letting the signal go into the red zone, as this can lead to distortion.

Headphone output. Most mixers have a headphone jack with an associated level control that's independent of the main output. This means you can turn down the output so the audience doesn't hear anything, yet listen to the various channels on headphones to make sure everything is working properly.

MONO VS. STEREO

In most venues, mono is preferable so everyone hears the same sound. With stereo, some people will be in the "sweet spot," while others will hear mostly the left or right channel. To create a mono mix with a stereo mixer, move all panpots to the center position.

THE IMPORTANCE OF "GAIN-STAGING"

This is the process of setting levels properly—not too high, which can lead to distortion, and not low, which can lead to hiss. There are three main places to adjust levels in a mixer; here's how to set them.

- ***Preamp gain.*** Set this for the maximum level short of distortion (i.e., just below where the clip indicator lights). It's acceptable if the clip indicator lights briefly and very rarely; otherwise, reduce the gain.
- ***Master output.*** This will usually be calibrated in dB. 0 dB indicates that it is neither adding gain nor attenuating the signal. Negative values indicate attenuation, and positive values indicate gain. When starting a mix, set it at around –3 dB. Mixers generally have output meters that show the overall level leaving the mixer.
- ***Channel faders.*** Set these so that during musical peaks, the output meters peak at around –3 dB. If you need to increase the level a bit as the venue fills up, no problem—simply turn the master output to 0, or even a little bit higher. If you need to go much higher than 0, turn up the GSR series speaker level controls.

MIXER MEETS COMPUTER

Some mixers, like the M-Audio NRV10, include a computer interface that allows you to send audio into a computer as well as to your speakers. If the computer is equipped with recording software, then it's possible to simultaneously record the live concert, conference, service, etc. Although computer-based recording is beyond the scope of this publication, it's important to realize that there's the potential to add recording to your live performance if you choose the correct mixer.

Fig. 19: The dark blue lines indicate signals going through a channel strip. The light blue lines show how signals come off the channel strip audio into knobs, which regulate the output going to the mixer's monitor bus. The yellow lines indicate a second send of sends, which feed the mixer's internal effects (e.g., reverb and the like).

Fig. 20: The signals from several channels, for example multiple mics from a choir, go to Aux Send bus 1. Returning its output to a mixer input (in this case, the Line In 7/8 channel) lets you use the Line In 7/8 fader to alter the level of all mics together, without changing the balance of the individual mics.

CHAPTER 6:

Signal Processing

There are several ways to improve the sound of your PA system, both on individual channels and the overall output, by using signal processors. Think of these as the "spices" that bring out the best in the signal sources feeding the PA. But to use them effectively, you need to understand how they work.

EQUALIZER TYPES

An equalizer (EQ) is a tone control that helps shape a sound's tonal quality. Thin voices can be made fuller, "muddy" sounds made clearer, and you can even use equalization to match your PA's sonic character to specific venues.

As mentioned previously, you'll usually find equalizers in mixer channel strips. However there are many different types, from simple to complex. Here's a summary.

Shelving EQ

This is like the bass or treble control on a typical hi-fi system. It gets its name because it creates a "shelf" in the frequency response (Fig. 21). A treble (or "high") shelving equalizer boosts or cuts high frequencies, whereas a bass (or "low") shelving equalizer boosts or cuts low frequencies. Even budget mixers usually include treble and bass shelving EQ.

Parametric EQ

The response is different compared to shelving equalizers, as it creates a peak (or notch) at a specific frequency (Fig. 22). A parametric EQ has three controls.

- ***Boost/cut.*** Also called Gain, this determines the amount of the peak (boost) or depth of the notch (cut).
- ***Frequency.*** This sets the frequency at which the boost or cut occurs.
- ***Bandwidth.*** Also called Q or resonance, this specifies the range of frequencies affected by the boost/cut control–from narrow to wide.

Quasi-Parametric EQ

This is like a standard parametric, but the bandwidth is fixed instead of variable.

Graphic EQ

This has multiple sliders–anywhere from three to dozens–with each slider determining the boost or cut amount at a specific, fixed frequency (Fig. 23). It gets its name because the sliders' physical positions show a rough approximation of the frequency response. Graphic equalizers are seldom part of a mixer's channel strip, but instead are included in the mixer's master controls to shape the overall output.

Multi-Stage EQ

Sophisticated mixers often combine multiple equalization stages. For example, a high-end mixer might have a *four-stage* EQ that combines a bass shelving EQ, treble shelving EQ, and two parametric stages. This allows for versatile equalization. Even relatively low-cost mixer channel strips may include three-stage equalizers with high and low shelving, and a parametric or quasi-parametric stage.

SHELVING EQ APPLICATIONS

Shelving EQs are best for broad, general changes. However, you need to be careful in situations involving microphones, as too much added bass or treble can lead to feedback.

- ***Audience compensation.*** People and clothing absorb sound, particularly high

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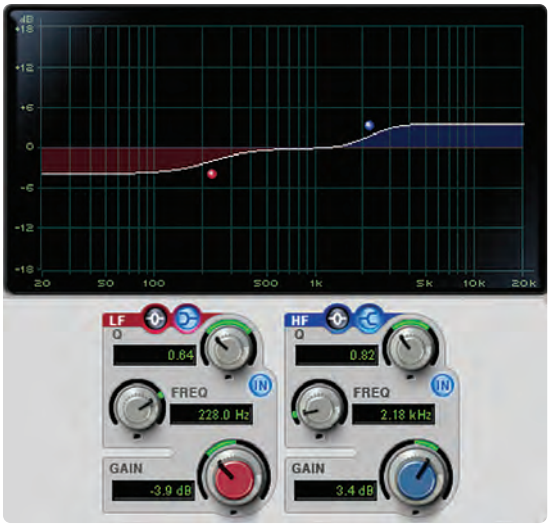

Fig. 21: Shelving filter response. There is a high-frequency boost, and low-frequency cut. The name "shelf" comes from the fact that after the response rises or falls, it levels off into a straight line.

frequencies. So as a room fills up, the overall sound might seem somewhat duller. A slight high-frequency boost at the output can help.

- ***DJ applications.*** DJ audiences typically expect more bass than, say, a conference or acoustic music concert. Boosting the bass results in a more crowd-pleasing sound.
- ***Making presenters sound more authoritative.*** Conferences and lectures usually don't require high volume levels, so you can get away with greater bass or treble boosts than you could compared to, say, a concert. Assuming any mics have wind filters to minimize "popping" (see Chapter 2), and their low-pass filters are enabled, you can make the presenter sound more authoritative by boosting the bass a little on his or her mixer channel strip for a fuller, bigger sound.
- ***Increasing a presenter's intelligibility.*** If people in the back row can't understand what a presenter is saying, boosting treble to some extent can help intelligibility.
- ***Fixing a "muddy" sound.*** If the PA's sound seems muddy, try reducing the bass at the output.
- ***Fixing a strident, or harsh, sound.*** Reducing treble at the output can give a warmer, less harsh timbre.

PARAMETRIC EQ APPLICATIONS

Parametric EQs are optimum for precise applications.

- ***Reducing hum.*** If there's AC power-related hum in a channel, dial in a frequency of 60 Hz (50 Hz in Europe). Set a narrow (sharp) bandwidth, and cut the gain to reduce the hum.
- ***Reducing feedback.*** It's a tricky to get this right, but parametric EQ can sometimes help reduce feedback. See Chapter 9 on Troubleshooting for more details.
- ***Separating "competing" sounds.*** For example, suppose a singer/songwriter is singing and playing guitar. Part of the guitar's frequency range will overlap with the vocal, thus competing with the vocal sound. A fairly broad cut on the guitar channel in the vocalist's range gives more space to the vocal. The exact frequency to cut depends on the singer; a bass or baritone can go below 100 Hz, while a high soprano can hit notes at around 1,200 Hz (with overtones in either case going higher). Thus, consider cutting the guitar starting around 500 Hz, then adjust up or down as required by the singer.

GRAPHIC EQ APPLICATIONS

- ***"Tuning" a room.*** A room is essentially a filter; a space with thick drapes, carpets, and wood chairs will absorb more sounds–particularly higher frequencies–than a concrete room with metal chairs, which will be extremely bright. There will also be resonances that boost some frequencies, and reduce others. By listening to a known music source, like a favorite CD, you can adjust the graphic equalizer's faders for the most accurate sound. This will change as you move around the room, so try for a good average setting rather than optimizing the sound for a specific "sweet spot."
- ***General timbre changes.*** You can also use a graphic equalizer to create the type of timbral changes described earlier for shelving EQ.

REVERB

Some PA mixers include reverberation to simulate the effect of being in a concert hall or large room. While this can sound impressive, most pros turn it off as the PA is already in a room, so adding reverb usually results in a less distinct sound. The same is true for effects like delay, echo, and "chorusing." Less is more!

LIMITER

A *limiter* is an electronic circuit that acts like a motor's governor—it won't let the sound exceed a certain level. If a mixer includes a limiter, it will usually be at the output (although some high-end mixers include limiters on individual channels).

A limiter will have an indicator that shows when limiting occurs, which means the signal level (and therefore the dynamic range) is being restricted. You don't want this to light very often; a limiter is more of a "safety valve." If this light illuminates a lot, you'll probably need to reduce the overall output level control. If it lights a lot on channel strips, then you'll likely need to reduce the amount of channel gain.

COMPRESSOR

A compressor also reduces dynamics but unlike a limiter, reduces peaks and amplifies lower-level signals to maintain a more constant level. The most important controls are *threshold* and *ratio*. Lower thresholds amplify low-level signals more, while higher ratios reduce high-level peaks more. While compressors can help with presenters at conferences where noise levels aren't too high, amplifying low-level signals with musical ensembles can encourage feedback. For most applications, leave compression off.

OTHER EFFECTS

The above are common effects for PA systems, and are intended mainly to optimize the sound. However, some mixers (like the M-Audio NRV10) include effects that are more properly considered "special effects." While it's doubtful you'd use these in a conference or worship situation, DJs can use special effects artistically

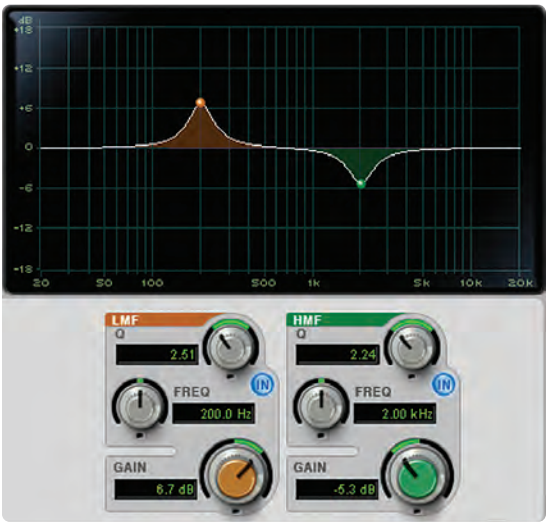

Fig. 22: Parametric filter response. A parametric filter can provide a peak or a notch in the frequency response.

to create new types of sounds, and theater groups can employ effects for tricks like making voices sound as if they're coming from a different dimension. Mixers may also feature built-in flanging (which imparts a "whooshing" character to the sound), tremolo (a rapid, cyclic fluctuation in level), delay (also called echo, which produces the "shouting 'hello' into a canyon" effect), and chorusing, which makes an individual instrument sound more like an ensemble.

Fig. 23: A graphic EQ gets its name because the sliders draw a "graph" of the frequency response.

CHAPTER 7:

Amplifying Musical Instruments

Compared to amplifying voice, the process of sending instruments through a PA system can be more complex. There are two main options for amplifying instruments:

- Mic acoustic sources, and feed the mic out into a mixer.
- Patch a direct electrical signal from electrical sound sources, such as electric guitar, synthesizer, electronic drums, and drum machines, etc. through patch cords into the PA system.

Let's consider miking first. As explained in the chapter on microphones, the microphone type and placement are the most important factors in sound quality. So, let's relate these to various miking situations.

MIC PLACEMENT FOR ACOUSTIC SOURCES

Vocals. While there are several recommended "best" ways for singers to use a mic (holding it away, holding it close, singing across it, singing above it, singing into it, or a combination of these as needed to emphasize or de-emphasize specific aspects of the song), every person's voice requires a different technique because every voice is different. For many voices, condenser mics sound best; for others, a dynamic mic gives the right sound. Also, refer to the chapter on Setup for tips on how to avoid feedback when miking vocalists.

Acoustic guitar. It may not be necessary to use a mic, as many acoustic guitars have pickups and onboard preamps that feed the output directly into a mixer. If that isn't an option, the mic of choice is generally a condenser type pointed between the sound hole and the bridge. Pointing directly at the sound hole can give a "boomy" effect, while pointing directly at the bridge often sounds "thin."

Harmony vocals. Mic placement tends to be further away from harmony singers than lead singers, especially if two or more singers are sharing a mic. This is also a situation where subtle angling can differentiate the background voices from the lead vocal.

Wind instruments. Use a dynamic mic to cope with the high sound pressure level, and back it off a foot or so from the instrument being miked. With sax, the mic is generally a few inches above the bell, and pointing straight at the sax.

Piano. Open up the lid and use two condenser mics, one pointing at the high strings, the other pointing toward the lower mid/bass strings. Make sure both are far enough away from the hammers to avoid picking up the "thunk" when they hit the strings.

Drums. There are several approaches to miking drums, from having a forest of mics covering every drum to a more minimalist approach. Less is often more, and many people mic drums successfully with just three or four mics: a dynamic mic for the kick (either just outside the head, or inside the drum), a condenser between the snare and high-hat to pick them up, an overhead condenser mic above the drummer's shoulder (pointing down toward the toms and angled away from the cymbals), and perhaps one or two room mics. However, optimum placement depends on the drum set, the desired prominence of different drum sounds in the mix, the drummer's individual setup, etc.

Guitar and bass amps. Many amps now include electrical direct outputs designed specifically for feeding mixers and PA systems. If you can, use these to keep your life simple! But if that's not possible, with a single-speaker cabinet, point a dynamic mic at the center of the speaker, a few inches away from the grille cloth, or move the mic more toward the edge of the speaker for a thinner, "tighter" sound. If the cabinet has multiple speakers, try miking just one. Also remember that when miked, a small, low-power amp can sound just as big as a stack of amps—without causing leakage problems into other mics.

ELECTRONIC INSTRUMENTS

This category includes synthesizers, electronic drums, stage pianos, drum machines, and the like. These instruments typically produce line-level, unbalanced signals, and can plug directly into a suitable mixer channel input. They often include onboard signal processors, so they get their "sound" before going into the PA mixer, and require no adjustments other than level.

Some players with extensive stage setups (e.g., a keyboard player with multiple keyboards, each with its own set of outputs) might also use a *submixer*. A submixer is conceptually like a standard mixer, but is usually designed to mix several line-level inputs, eliminating the need for mic preamps, extensive EQ, and so on. Once the player achieves the desired instrumental balance with the submixer, then its master stereo output can be sent to the main PA mixer.

ELECTRIC GUITAR AND BASS "DIRECT" SETUPS

Carting around a big amp and lots of signal processors can get pretty tiresome, so many guitarists downsize their setups to compact digital processors. These digital processors emulate the sounds of effects, amps, and even particular mikings of those amps.

A good example is the Avid® Eleven® Rack. It includes a variety of amp, effects, and mic models, making it possible to get a complete guitar sound in a single, portable box. (In addition to live performance applications, Eleven Rack doubles as a computer interface for Pro Tools LE® recording software.) Setup is easy, as you simply plug the guitar into the Eleven Rack input, and plug the outputs into your PA system.

Another option is loading amp modeling and effects software (such as Avid Eleven) into a laptop computer or even one of the more powerful netbooks. This requires an audio interface that connects to the computer's USB port, FireWire port, or card slot (e.g., PCMCIA or ExpressCard) to provide physical audio inputs and outputs. Like Eleven Rack, the guitar plugs into the input, and the outputs go to the main PA mixer. Note that the audio interface must have an input that's designed specifically to accommodate guitar.

Setups for electric bass are similar.

SINGER/SONGWRITER SETUP

You can insert a microphone or line-level input into a powered speaker such as the GSR10 and GSR12, so having two of these speakers is ideal for a simple, highly portable singer/songwriter setup. The mic goes into one speaker's mic input, while the output from an electric piano, synthesizer, or preamp output from an acoustic guitar can go into the other speaker's line-level input. This eliminates the need for a mixer, and each can have its level adjusted independently. Note: This is a situation where the GSR tone options can make a huge difference to the overall sound—try them all.

CHAPTER 8:

Sound Check

Now that everything is set up, it's time to test out the system and make sure everything works and sounds as expected.

"RINGING OUT" THE SYSTEM

The optimum situation is to check out the sound with the performers or presenters who will be playing through the system. However, this isn't always possible, so you may need to set up hours beforehand when the rest of the people aren't around.

Many sound engineers bring a portable CD player or other portable music player with music they know very well, like a favorite CD, and play that through the system after setup to get a rough idea of the system's performance. You can use this music to check for several sonic qualities.

Coverage. While the music is playing, walk around the room. Ideally, the sound should be identical wherever you are, but this is seldom the case. You may need to change the speaker angle and/or height somewhat to obtain the most uniform coverage.

With the M-Audio GSR series speakers, it's possible to daisy-chain up to three speaker cabinets (Fig. 24). This provides many additional speaker placement options. For example, you may get better coverage by stacking a speaker on top of another speaker, or by placing the two side by side—experiment to determine which is best, as acoustics vary considerably from room to room.

Placement is particularly important for high frequencies. As you walk around the room, make sure that everyone has a straight line of sight with the speakers (Fig. 25). If not, angling multiple speakers somewhat outward or inward may give better coverage for a larger segment of the audience.

Fig. 24: The GSR series speakers include a "Thru" jack that can pass the input signal from one GSR speaker to another GSR speaker's input.

If the bass from the subwoofer seems weak in some spots, place the back of the subwoofer against the wall, or even at the corner of two walls. Any sound coming from the subwoofer's back bounces off the wall, reinforcing the main bass signal. Although not necessarily a common technique, some sound engineers diffuse the bass sound more by pointing the subwoofer front toward a corner. This can increase the apparent amount of bass, as well as cause the wall to vibrate, which creates additional bass.

Tone. As mentioned previously, the tone can change dramatically depending on the number of people in a room. Therefore, when "ringing out" a PA (a term pros use to describe the sound check process), you mainly need to analyze the room's character so that you can make adjustments later. For example, if the room has really hard surfaces, the PA may sound a little bright during sound check but if the room fills up, then some of the highs will be absorbed. It's also worth trying the different tone options in the GSR series speakers (Normal, Hi-Fi, DJ, and Voice—see Fig. 26) as you may find that one option gives a better overall sound for a particular room than another.

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Fig. 25: Pole-mounting a speaker on top of the woofer can often provide better high-frequency coverage for your audience.

If a room has a lot of soft surfaces–curtains, plush seats, a packed house, etc.–then you may find that the Hi-Fi setting gives a more balanced sound than the Normal setting.

Veterans will tell you experimentation is key, and it is. However, be careful. Switching from one tone to another may sound great because it's a change that sounds fresh, not necessarily because it's better. When you make any change, live with it for a while, then switch back to how it was originally. Do this a couple times to choose the "winner." Then if necessary, you may want to try yet another change and A-B the new sound with the winner of the previous A-B comparison.

ADJUSTING FOR ROOM RESONANCES

As mentioned in the Signal Processing chapter, some mixers have graphic equalizers at the outputs. These make it easier to "tune" a room, but proper adjustment requires some experience.

When you listen back to a CD you know well, analyze whether any parts of the frequency spectrum sound different than normal. For example, if the upper midrange sounds harsh, there may be a room resonance that accents this part of the spectrum. Try bringing down a graphic EQ slider in the 3–5 kHz range, and listen for whether that makes a smoother overall response. Conversely, if there are lots of soft surfaces, then part of the midrange may sound attenuated, which can make presenters sound less distinct. In this case, a slight upper-midrange boost might solve the problem.

CHECKING SOUND LEVELS

Some areas have ordinances that govern the maximum sound level for particular venues. To make sure you are in compliance, use a *sound level meter.* These devices have a calibrated microphone and meter that shows the level being produced at the location where you're checking levels. Sound level meters are available from electronics stores like Radio Shack, although there are also iPhone and iPod Touch apps that provide sound level metering.

Fig. 26: The GSR series' Tone switch tailors the speaker's response to different environments.

CHAPTER 9:

TROUBLESHOOTING

Once you learn the basics of running a PA, you'll find the process to be relatively trouble-free. However, there will be times when you encounter problems, and these tips should help get you through most of them.

SOLVING "OUT OF PHASE" PROBLEMS

Phasing problems typically occur when you're using two mics on a single sound source (e.g., miking a piano), and each mic picks up a different portion of the sound wave. For example, if one picks up the wave's peak while another picks up the wave's trough, the sounds will cancel to some degree (see Fig. 27; note that we're simplifying for the sake of illustration). Although each mic might sound fine by itself, when combined the sound can seem "hollow" or "thin." The sonic change when a jet flies overhead comes from phase changes as you hear different reflections.

To check for phase issues, your mixer will probably have a phase switch that reverses the mic preamp's phase. If flipping the phase switch on *one* of two mics creates a bigger, "fatter" sound when they're mixed together, then leave the phase flipped. If there's no phase switch, moving the mic a few inches closer or further away will often reduce the problem. Your goal is the best possible sound when the two mic signals are mixed together.

DEALING WITH FEEDBACK

Feedback is that annoying howl that occurs when a signal from the speaker enters the mic, gets amplified, comes out of the speaker again, hits the mic again, gets amplified some more, and so on—creating what's called a *feedback loop.* As mentioned in Chapter 2, there's no "magic bullet" for fixing feedback because the primary cause could be mic placement, speaker placement, mixer control settings, equalization, or even how a vocalist holds the mic. Here are tips on reducing the potential for feedback.

Use directional mics; never use omni-directional mics. The mics should point away from the loudspeakers and toward the sound source.

Set up mics behind the main speakers. This reduces the chance of sound from the speakers entering the mics (Fig. 28). The downside is that this also makes it difficult for vocalists or lecturers to hear what they sound like through the speakers, which is why monitor speakers or reference earphones are commonly used in PA setups (see Chapter 4).

Avoid "swallowing" the mic. Vocalists who hold the mic very close often run a greater risk

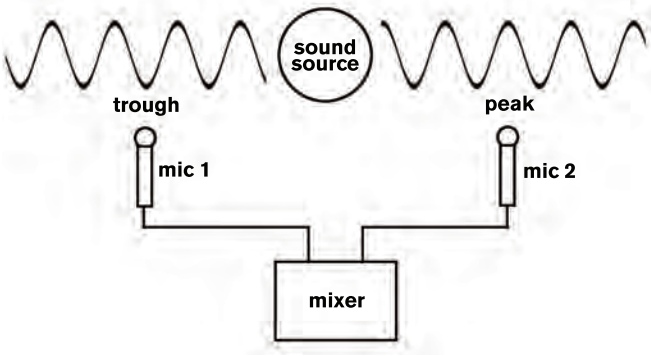

Fig. 27: Sound waves emanate from the sound source, and two mics are positioned to pick up these sound waves. However, one picks up the signal's trough while the other picks up the signal's peak, causing the combined signal to cancel to some degree.

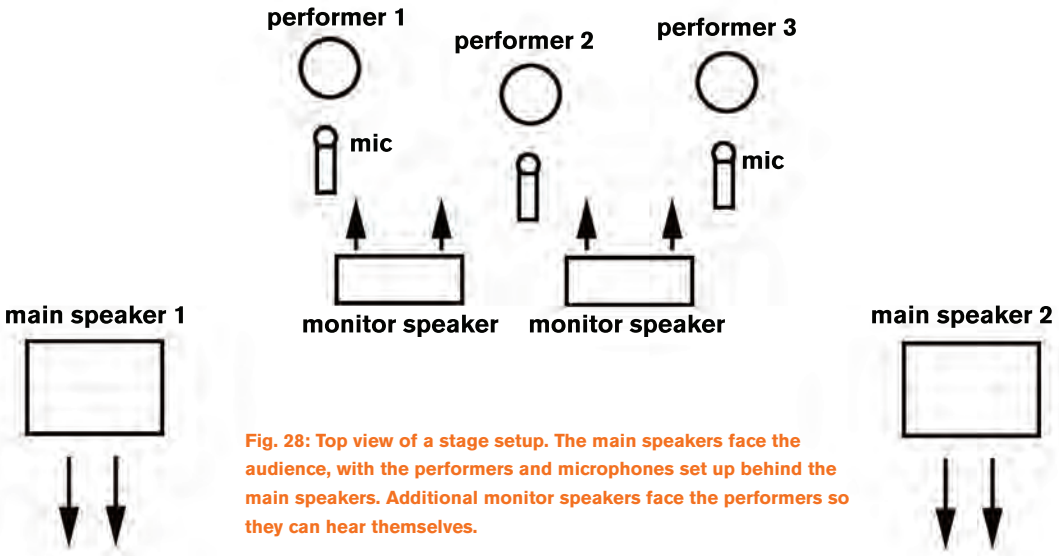

Fig. 28: Top view of a stage setup. The main speakers face the audience, with the performers and microphones set up behind the main speakers. Additional monitor speakers face the performers so they can hear themselves.

of feedback than if the mic is an inch or two away. This is because the "proximity effect" increases the amount of bass, which encourages feedback in the bass frequencies.

Try adjusting the equalization. When feedback starts, it will be at a specific frequency. Solo each channel to find out which one is triggering the feedback. Set the EQ for a steep notch, then sweep the frequency control until you find the frequency that minimizes the feedback. While this may affect the timbre of the sound going through that channel, the reduced feedback may be worth it. You may even be able to find another feedback frequency on another channel, reduce that, and have even more level before feedback occurs. If the feedback seems to occur in all channels, you may need to adjust EQ at the mixer output. A graphic equalizer will likely not allow for fine enough adjustment compared to a parametric equalizer, but if you're lucky, the feedback frequency might correspond to a graphic EQ filter frequency.

MINIMIZING DISTORTION

Distortion can occur in several places in the system (see the section on Gain-Staging in Chapter 5). First, check to see if any clipping indicators light when the distortion occurs. If the indicator relates to a preamp, reduce the preamp gain. If the indicator relates to the overall system (e.g., the clipping indicator on a power amp lights), then you need to reduce the master volume or all the faders feeding the master output. Granted the system won't be as loud, but a softer sound is a better option than a distorted one.

Second, distortion may be occurring in the sound source itself. A damaged mic, for example (e.g., it's been dropped a couple times) may have a fuzzy, distorted sound. Distortion—intentional or unintentional—can also happen in something like an instrument amplifier.

REDUCING HUMS AND BUZZES

Hums and buzzes are generally from one of two sources.

The first is lighting dimmers or fluorescent lights that generate interference. Mic preamps, cables, guitar pickups, and other sensitive or high-gain circuits can easily pick up this interference. It's simple to test for if this is the problem: Turn off the lights and see if the problem goes away. If the problem relates to dimmers, turning the dimmer all the way up or down will usually solve the problem. Incandescent bulbs that don't use dimmers are least problematic.

Some appliances can also generate interference. If lights aren't the problem, check whether unplugging appliances on the same circuit reduces any buzzes. In many cases, purchasing a line filter and making it part of your PA setup is good insurance against these types of buzzes and hums. However, any filters need to be rated for sufficient power to handle a PA system–the kind for home entertainment systems won't be up to the task.

The second problem involves ground loops. A ground loop means there is more than one ground path available to a device. In Fig. 29, one path goes from device A to ground via the AC power cord's ground terminal, but A also sees a path to ground through the shielded cable and AC ground of device B. Because ground wires have some resistance (the electronic equivalent of friction), there can be a voltage difference between the two ground lines, thus causing small amounts of current to flow through ground. This signal may get induced into the hot conductor. The loop can also act like an antenna for hum and radio frequencies. Furthermore, many components in a circuit connect to ground. If that ground is "dirty," the circuit might pick up the noise. Ground loops cause the most problems with high-gain circuits like mic preamps, as massive amplification of even a couple millivolts of noise can be objectionable.

One supposed "fix" is to "lift" the AC ground by plugging a three-wire cord into a 3-to-2 adapter. However, this is *definitely not recommended* as it eliminates the safety protection afforded by a grounded chassis. Don't do it.

A better solution is to plug all equipment into the same grounded AC source, which attaches all ground leads to a single ground point (for example, a barrier strip that feeds an AC outlet through a short cord). However, make sure that the AC source is not overloaded and is rated properly to handle the gear plugged into it.

A more expensive, but almost always effective, solution is isolating the piece of gear where the ground loop occurs by patching the audio through a *1:1 audio isolation transformer.* This interrupts the ground connection, but still allows the signal to pass.

Another option–although the most expensive one–is using an *AC power isolation transformer* to power your gear. This can also provide ancillary benefits: clean up noise from the AC line, reduce spikes and transients, and provide performance almost equal to that of a separate AC line.

If you have a lot of microprocessor-controlled gear and less-than-ideal AC power, adding isolation transformers can solve various AC-related problems *and* get rid of ground loops. If you just have a simple ground loop problem, then patching in audio isolation transformers in specific audio lines may be all you need.

Granted, solving any hum and buzz problems may sound daunting. Fortunately, in most cases you won't have to worry about these issues, particularly in smaller setups.

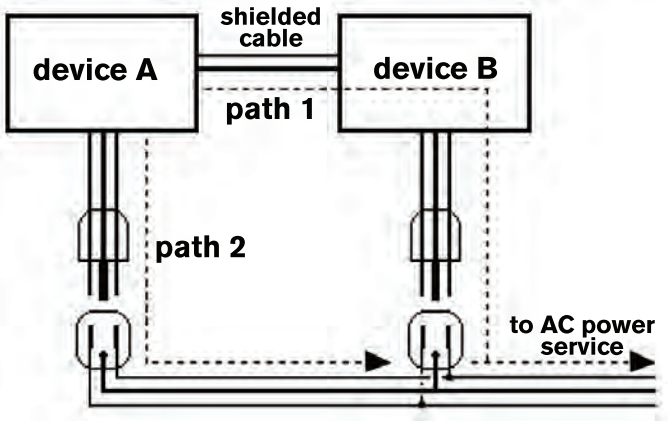

Fig. 29: When a device can take two separate paths to ground, this creates a ground loop. Whether it matters or not depends on several factors, such as the "dirtiness" of AC power, and also, whether the device is high-gain or not.

CHAPTER 10:

GLOSSARY

Amplifier
A device with an input that receives weak electrical signals, makes them stronger, then sends them to an output suitable for driving loudspeakers. Note that while amplifiers are available as separate devices, they are often built into speaker cabinets or mixers as a matter of convenience.

Auxiliary Send
See *Send Bus.*

Cable
Wire that has connectors on both ends, and is designed to interconnect pieces of gear.

Clipping
The phenomenon that occurs when an input signal exceeds the headroom (dynamic range) that a system can handle, thus distorting the waveform because it cannot be reproduced accurately.

Clipping Indicator
A light, typically found on mixer channels and amplifier outputs, that indicates when a signal exceeds the maximum available headroom. When a clipping indicator lights, you need to turn down the level in the system (either at individual mixer channels, or the master level if clipping occurs in the power amplifier).

Distortion
What happens when a waveform cannot be reproduced properly for one reason or another (e.g., inadequate power-handling capacity).

Equalizer
A device that alters the signal level at specific frequencies, for example, making the higher frequencies softer to prevent a shrill or "tinny" sound, or increasing the level of lower frequencies to give more bass.

Fader
A volume control, typically with a linear path rather than a rotary one.

Input Channel
The part of a mixer dedicated to a particular input. It will typically have an input and fader to regulate the volume, but can also include equalization, effects, meters, clipping indicators, and other elements designed to enhance the mixing process.

Jack
A connector for cables that receive or send signals among devices. "Female" jacks receive signals from "male" plugs.

Level
Another term for volume or loudness. For example, increasing a mixer channel's level control will increase the volume of the signal going into that channel.

Line Level
A relatively strong signal level, such as what emanates from electronic musical instruments, CD players, MP3 players, and the like.

Mic Level
A relatively weak signal level, specifically what emanates from a microphone.

Microphone
A transducer that converts acoustical energy to electrical energy.

Master Volume
The control on a mixer that regulates the overall level of the sum of all channels.

Mixer
A device that combines multiple signal sources (typically microphones and instrument outputs) and sends a combined signal into an amplifier.

They include controls to adjust the relative level of each signal source in order to achieve a proper audio balance. Mixers are specified as having a certain number of channels—e.g., a 4-channel mixer can accept four different input sources, while a 16-channel mixer can accept 16 different input sources.

Monitor Speakers
Speakers placed in front of performers to let them hear themselves. These are not intended to be heard by the audience, and may be smaller than the main speakers. Common monitor speakers are *wedges,* so called because they sit on the floor, and project sound upward and toward the performer.

Panpot
With stereo setups, this is a per-channel mixer control that determines whether a signal will be heard mostly from the left side, right side, center, or for that matter, anywhere in the stereo field. When PA setups are set up in mono, the panpot is left at the center position.

Phantom Power
Most condenser microphones require a power source in order to function correctly. Rather than have a power source inside the mic or in a separate unit that connects to the mic, a phantom power system sends power (typically +48 volts) along the microphone cable. Dynamic microphones do not need phantom power, but if enabled accidentally, will seldom suffer damage from receiving phantom power.

Phone Connector
A type of connector for balanced or unbalanced lines, typically found in semi-pro and pro audio gear. It is also called a 1/4" phone connector because the barrel of the plug, or hole of the jack, has a 1/4" diameter.

Phono Connector
A type of connector for unbalanced lines typically found on consumer hi-fi gear. Also called an RCA connector.

Powered Mixer
A mixer that contains a built-in power amplifier, and therefore can drive speaker cabinets that don't include built-in amplifiers directly.

Powered Speaker
A speaker cabinet/system that includes a built-in amplifier. This approach allows matching the amplifier and speaker.

Proximity Effect
A phenomenon generally associated with dynamic cardioid mics where as the vocalist gets closer to the mic, the apparent bass response rises.

Return
A mixer input that receives the output from a signal processor fed by an auxiliary or send bus.

Send Bus
A separate submix from a mixer that combines various amounts of signals from different channels. This is used for two main purposes. One is to provide a mix separate from the main mix that goes to the performers (either to headphones or monitor speakers), as this mix may need to be different from what the audience hears (e.g., vocalists might want to hear themselves louder than other instruments). The other is to add signal processing to channels in various degrees by feeding a signal processor. For example, you might want to send a lot of vocal signal to a reverb unit to add a sense of spaciousness, but send only a little bit of the guitar and none of the drums. The knob that determines the amount of signal that's sent is called a send control or aux send control.

XLR Connector
A three-pin connector used with balanced lines and found in pro audio gear.

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Please visit m-audio.com for complete performance specifications as well as detailed information regarding compatibility.

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